IMPULSE RESPONSE MEASUREMENT OF ACOUSTIC SPACES:

A PROTEAN APPROACH FOR CONCENTRIC APPLICATIONS IN CONVOLUTION REVERB PROCESSING AND ACOUSTICAL ANALYSIS

A CREATIVE PROJECT

SUBMITTED TO THE GRADUATE SCHOOL

IN PARTIAL FULFILLMENT OF THE REQUIREMENTS

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MASTER OF MUSIC

BY

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Part 1: Topic Overview

Introduction

Musicians often exhibit a certain amount of preoccupation with the acoustic parameters of the rooms in which their art is realized. From a mechanical perspective, a typical musical performance occurs in three essential phases: 1) the generation of sound energy (performance), 2) the propagation of sound through a medium, and 3) the listener's reception of and subsequent decoding and interpretation of the sound (music perception and psychoacoustics). Performers and listeners are often preoccupied with the first step, which concerns itself with the technical proficiency of a musical performance and the quality of the sounding instruments. However, many composers, musicians, and audience members often underestimate the degree of influence exerted during the intermediary second stage,[1] in which the acoustic characteristics of the performance venue modulate (transform) the source signal as it travels from performer to listener.[2] Ultimately, the performance space itself is responsible for successful transmission, representation, and perception of a sound image. An ideal venue contributes desirable effects of reverberation and spectral presentation (timbre) that offers a flattering representation of the sound source, while undesirable acoustics can lead to an unforgivable presentation lacking in clarity, tone, and spatial impression.

This project primarily concerns itself with the intermediary stage of acoustic propagation, with an emphasis on the measurement and application of concert hall acoustics. Sound can be measured and described as a combination of three essential parameters: frequency (pitch),

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[1] Daniel J. Levitin, *This is Your Brain on Music: The Science of a Human Obsession* (New York: Plume, 2007), 16.

[2] Per Rubak and Lars G. Johansen, "Coloration in Natural and Artificial Room Impulse Responses" (presentation, 23rd International AES Conference, Copenhagen, Denmark, May 23-25, 2003).

amplitude (loudness), and duration (time, as well as phase). A rudimentary definition of music is the active, intentional manipulation of frequency and amplitude over time. Musicians actively manipulate these parameters during performance, which correlates to the familiar notions of pitch, dynamics, and timbre. However, the direct sound radiated by an instrument or vocalist is rarely received by the listener without significant alteration due to the hall's influence on the frequencies and amplitudes generated by the performer; the product is transformed during delivery.[3] The listener might perceive these differences as effects on timbre, loudness, balance, spatial image, or any number of qualitative perceptual paradigms. However, it is possible to define these subjective phenomena as the result of measurable, *quantifiable* parameters whose configuration result in the unique characteristics or "sound" or "color" of a particular hall.[4] The measurement of a hall's frequency response over time is known as its *impulse response*.[5] Quantitative analysis of an impulse response allows one to predict a hall's influence on a listener's perception of frequency, amplitude, and their changes over time.

Furthermore, the data yielded during the process of impulse response measurement have several notable applications. Impulse response measurements are indispensible for quantifiable analysis of a space's acoustical characteristics, such as reverberation, resonance, and diffusion. Impulse responses are also readily usable to recording engineers, music producers, composers, and those involved in cinematographic audio post-production.

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[3] Arthur H. Benade, "From Instrument to Ear in a Room: Direct or via Recording," *Journal of the Audio Engineering Society* 33, no. 4 (1985): 218-233.

[4] Rubak, "Coloration in Natural and Artificial Room Impulse Responses," 1-2.

[5] Curtis Roads et al., *The Computer Music Tutorial* (Cambridge, MA: The MIT Press, 1996), 474-476.

Convolution reverb is a process in which a pre-recorded impulse response of a space can be used to simulate its reverberant and spatial characteristics at any time.[6] Any live or pre-recorded audio signal can be processed to give the impression as if it were originating and propagating throughout the simulated space. Thus, the spatial and spectral information of a space – its acoustic fingerprint – can be digitally recorded and recreated at any time. If a performer does not have the luxury of performing on an actual stage, the illusion of one can be generated in a recording before it is distributed. In essence, the acoustics of the hall become portable, making the venue available for recording even when the physical space itself is not.

Procedure Overview

There are many computer programs and specialized hardware processors that employ convolution reverb and impulse response analysis. However, all of them operate using the same basic premise, requiring two types of audio signals. The first is the raw or "dry" audio signal, such as music or speech. This source signal can be either prerecorded audio or a continuous, analogous feed from a live microphone. The second type of audio signal is the *impulse response* of the desired performance venue. An impulse response is a digitally-recorded audio file that contains information on the *envelope* (change in amplitude and phase over time) of each audible frequency as propagated through the measured medium. For example, the impulse response of a "bright" concert hall would boost the amplitude of high frequencies, whereas a "dark" concert hall would attenuate the upper reaches of the audible spectrum.[7] Similarly, a "wet" concert hall with long reverberation time would extend the duration of each frequency's envelope, while a "dry" space would feature short durations and little reinforcement or resonance.

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[6] Ibid., 426-428.

[7] Thomas Rossing, Paul Wheeler, and Richard Moore, *The Science of Sound*, 3rd ed. (San Francisco: Addison-Wesley, 2002), 535-538.

Furthermore, impulse responses are not exclusive to concert halls, cathedrals, or auditoriums. Any acoustic space can be measured and synthesized utilizing convolution reverb, so long as sound waves are able to propagate and experience resonance. Parking garages, stairwells, showers, and racquetball courts are recognizable examples of highly-resonate spaces in which prolonged acoustic reverberation is an unintended byproduct of their respective architectures, which are engineered for utility rather than acoustic perfection. In film and television production, audio engineers capture the impulse responses of wherever the drama takes place – offices, elevators, caves, etc. By recording speech in isolation or post-production and utilizing convolution reverb, an audio engineer is able to make speech intelligible while giving the audience the subconscious illusion that the dialogue on-screen sounds as if it were naturally occurring in the environment. Moreover, it is possible to measure the impulse response of physical objects that exhibit resonance, such as springs, metallic plates, piano soundboards, and garbage cans, often through the use of contact microphones or magnetic pickups.

Audio files of impulse responses are not intended for recreational listening. Regardless of the space or hardware being measured, the impulse response waveform resembles (and often sounds like) a gunshot, with variations in timbre and length of decay. There is no single universal technique for recording impulse responses, though their approaches can be classified into four categories: sine sweep method, transient response, maximum-length sequence (MLS), and finite impulse response using the Fourier Transform. "Transient response" measurements usually involve the use of a starter pistol, balloon pop, or another sound source that is able to generate much of the entire spectrum in an instant and radiate sound omnidirectionally. The burst and subsequent reverberant decay is recorded by a microphone, and the resultant audio signal is a decent approximation of the room's impulse response. However, it is difficult to find a source that is able to accurately generate each possible frequency with equal loudness. The other three methods (sine sweep, MLS, and transfer function) utilize complex algorithms to construct an impulse response, but do not do so in real-time.

This project employs the sine sweep method, which provides the most accurate results.[8] The logarithmic sine sweep involves playback using a loudspeaker and microphone. The loudspeaker "auditions" all audible frequencies by generating a sine wave, the mathematically simplest possible waveform, which lacks harmonics or overtones. The speaker plays a sine wave at all frequencies in the specified sampling rate, with an upper limit that should exceed the frequency range audible to human ears (20 – 20,000 Hz). The sine wave oscillator logarithmically sweeps (vis-à-vis *portamento*) from the lower to the upper limit of the sampled range. A microphone placed within the room records the playback, identified as the *sweep response*. The computer then deconvolves the original sine sweep with the sweep response, which eliminates the common element (the direct sine sweep with no reverb or attenuation). The resultant audio file contains only the room's frequency response and decay for each frequency, and utilizes the Fourier Transform to eliminate the time domain of the original sweep. This deconvolved sweep response is the room's impulse response, which can then be convolved with any incoming signal to recreate the effects of the room's reverberant qualities.[9] The entire procedure is summarized in **Figure 1**:

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[8] Angelo Farina and Regev Ayalon, "Recording Concert Hall Acoustics for Posterity," (presentation, 24th International AES Conference, Banff, Alberta, Canada, June 26-28, 2003).

[9] Thomas Rossing, Paul Wheeler, and Richard Moore, *The Science of Sound*, 3rd ed. (San Francisco: Addison-Wesley, 2002), 643-651.

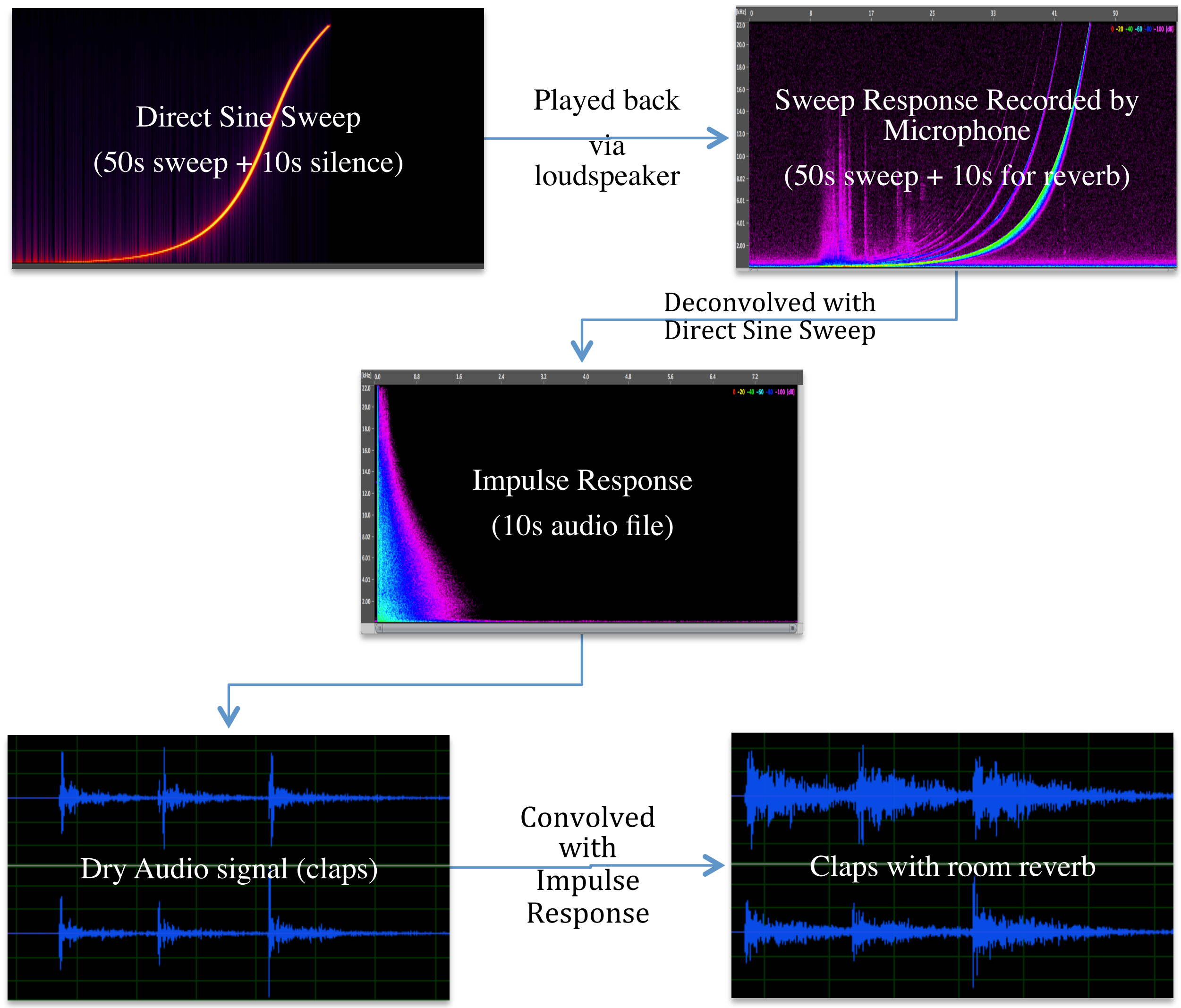

Figure 1: Impulse Response recording and application cycle

In order to maximize potential usability, this project will employ the impulse response recording procedures employed by three industry-standard convolution reverbs: Altiverb (manufactured by AudioEase), IR-1 and IR-360 (manufactured by Waves), and Space Designer, a tool included in Apple's Logic Pro X. Each of these programs are used by amateurs and professionals alike, and offer companion libraries of impulse responses recorded at esteemed halls, cathedrals, churches, and other performance venues from the entire world. By following the impulse response procedures outlined for each program, it is possible to create reverb plugins that are readily usable to audio engineers abroad. Furthermore, the impulse responses used by all three programs are compatible for further analysis and manipulation in programs such as Smaart (by Rational Acoustics) and MaxMSP (by Cycling '74). All three of the proposed formats are also compatible with the microphone techniques used in this procedure (Mono/Stereo Omni, ORTF and B-Format).

Microphone choice and configuration is a vital component of impulse response recording. The most basic configuration of the sine sweep method requires a single loudspeaker and s single microphone, usually with an omnidirectional polar pattern. However, the resultant impulse response is only applicable to a monophonic sound source and audio output. While a single sound source is applicable to vocal or instrumental soloists, it is an inadequate representation of wider ensembles, such as a symphonic orchestra or choir. By using multiple speakers, it is possible to approximate ensembles of variable widths. Furthermore, recording sine sweeps from speakers located in the rear of the hall allows one to cross the threshold from stereophonic simulation of the hall to full surround sound capability. This can be used to approximate performers standing *around* the audience in addition to those on stage. Surround sound impulse responses are potentially useful for those mixing audio for DVD, Blu-Ray, or any other format that supports surround sound. Electroacoustic composers can also use these impulses to binaurally simulate surround sound in a hall, which is a useful predictive mixing tool due to the relative difficulty of accessing a surround-sound system in a world-class concert hall.

Part 2: Description of Equipment

Microphones

In an attempt to maximize versatility, this project employs a variety of microphones, simultaneously capturing numerous perspectives of each sweep. The simplest microphone used in this experiment is the Neumann KM183, a small-condenser microphone with a notably neutral frequency response and omnidirectional pickup pattern. When placed at the point of capture, it is equally responsive to sounds arriving from all direction. The Neumann KM183 is placed in the center of the hall and records a single monaural track.

Another common microphone technique involves two omnidirectional microphones with a variable amount of space in between, sometimes known as an AB pair.[10] Although sound pressure arriving at each microphone is nearly equivalent, a stereo image is generated by capitalizing on the difference in arrival time. A sound equidistant from both omnidirectional microphones will appear to be in the center of the apparent stereo image. However, if a sound source is closer to one microphone or another, the nearest microphone will capture it first, causing the stereo image to pan left or right accordingly. This project utilizes two Sennheiser MKH80 microphones (with omnidirectional polar patterns) hanging above the first row of the balcony, which provides a realistic representation of an audience member perspective seated in the balcony.

The other two microphone configurations employed in this project are more complicated, and are discussed in detail below.

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[10] Rossing, *The Science of Sound*, 576-577.

Schoeps MSTC 64 U (ORTF)

This project employs two very special microphones to record impulse responses. The first is the Schoeps MSTC 64 U, a stereo microphone employing the ORTF configuration. The ORTF configuration is named after the *Office de Radiodiffusion Télévision Française* (French Radio and Television Organization), where it was invented in the 1960s.[11] The ORTF configuration features two near-coincident cardioid microphones, spaced 17 cm apart, spread 110° apart so that both microphones are off-axis from the sound source. The distance is approximately the same as that between two ears on a human head, and the angle approximates the directionality caused by the shape of the outer ear (*pinna*). The ORTF configuration gives an accurate stereo image, and its reliable, predictable behavior have made it an international standard for stereophonic recording. By combining aspects of differential time-of-arrival (caused by the space between the microphones) and intensity difference (caused by the directional bias of the microphones, which accepts on-axis sound and attenuates off-axis sound), it emulates both mechanisms that allow listeners to perceive and localize the apparent origin or direction of incoming sounds. The Schoeps ORTF microphone has a very clean, natural frequency response that equally presents the entire audible spectrum without notable biases.

SoundField MKV (Ambisonic)

The second microphone used in this project is the SoundField MKV, which employs principles of *Ambisonics* to record and recreate full three-dimensional surround sound. Although it first appears to be only a single microphone, it actually contains four discrete coincident capsules: Left Front Up, Left Back Down, Right Front Down, and Right Back Up, arranged in a

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[11] Stanley Lipshitz, "Stereo Microphone Techniques: Are the Purists Wrong?," *Journal of the Audio Engineering Society* 34, no. 9 (1986): 716-744.

tetrahedral formation. The LF and RB microphones are angled upward, while the RF and LB microphones are aimed downward. Instead of a traditional 3-pin XLR cable, the SoundField MkV features a special proprietary 10-pin cable that allows the transduced signals from the four capsules to be transferred using a single cable. The feeds from the individual capsules are not useful on their own; they require processing by a dedicated *decoder,* which converts the signals into more usable formats, such as mono, stereo, or B-Format (discussed below).

Overview of Ambisonic Technology (B-Format)

An Ambisonic microphone can be used to simulate any microphone of any polarity, pointed in any direction in all three dimensions; the only thing it cannot adequately simulate are microphones in different locations.[12] Although the physical microphone itself does not move, the controls on the decoder allow one to "aim" the microphone left, right, up, down, and even *zoom* in towards a sound source by simulating changes in microphone directionality and axial sensitivity. The decoder accomplishes this by converting the raw signal from the Ambisonic microphone (called A-Format) into the much more useful B-Format, which employs four discrete channels. Whereas a typical stereo configuration has two main outputs (left and right), B-Format features four signals, designated *W, X, Y,* and *Z*. The *W*-signal corresponds to sound pressure arriving from all directions, and is equivalent to a single omnidirectional microphone. The *X, Y,* and *Z* signals each correspond to figure-of-eight (bidirectional) polar patterns that align with axes of a three-dimensional coordinate plane.[13] The bidirectional *X*-axis is pointed toward the sound source, or toward (or away from) the stage from an audience member's perspective.

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[12] Hugh Robjohns, "You Are Surrounded: Surround Sound Explained – Part 3," *Sound On Sound*, October 2001, http://www.soundonsound.com/sos/oct01/articles/surroundsound3.asp.

[13] SoundField Research Ltd., *The SoundField MKV Microphone: User's Guide* (England: SoundField Research Ltd., n.d.).

The *Y*-axis corresponds to a listener's left and right, and the *Z*-axis corresponds to the plane between the ceiling and the floor. By manipulating signal phase and amplitude, the Ambisonic decoder is able to approximate directionality in a full three-dimensional field.

Ambisonic technology is especially valuable in post-production. By recording all four B-format channels, the same signal can be processed and re-decoded at any time; it is possible to explore all three dimensions *after* the recording has occurred. In his article "You Are Surrounded," Hugh Robjohns describes post-production of a B-Format recording made at the wedding of Prince Charles and Princess Diana.[14] Although the performance was recorded using only a single SoundField microphone it was possible to focus and zoom in on the lead vocalist, any instrument in the orchestra, or even individual audience members long after the nuptials had ended. Such *post hoc* flexibility allows one to re-balance or remix an entire performance without resorting to overdubs or exhaustive processing. A recording engineer might ruminate and toil over which stereo microphone technique to use, such as XY, Mid-Side, or Blumlein; the SoundField microphone allows one to choose which configuration *after* the concert has ended, making it easy to compare which configuration might have yielded the most desirable results. Furthermore, B-Format signals are compatible with any number of output channels. The signal is directly compatible with mono and stereo formats, and further decoding allows one to export a surround-compatible image for quadrophonic, 5.1, 6.1, 7.1, 8.1 and beyond – if future surround formats require additional channels, B-format signals recorded in the present are ready for whatever the future may bring.

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[14] Hugh Robjohns, "You Are Surrounded: Surround Sound Explained – Part 3," *Sound On Sound*, October 2001, http://www.soundonsound.com/sos/oct01/articles/surroundsound3.asp.

Playback/Recording System

Presonus StudioLive 24.4.2 Mixing Console

The Presonus StudioLive 24.4.2 Mixing Console is used to direct all incoming audio signals from the microphones and outgoing signals to the loudspeakers. It also serves as an audio interface connected to the researcher's computer, allowing digital recording of audio files in Pro Tools and Logic Pro X. Recording Impulse Responses requires the cleanest possible signal path, with the fewest possible intermediary connections possible. The console was placed on a table in the vestibule, so that most of the microphones could be plugged directly into the preamplifiers without needing to be routed through the hall's patch bay. However, due to limitations of XLR cable length and availability, a compromise was necessary, so the loudspeakers and balcony microphones utilized the patch bay while the most sensitively-positioned microphones were able to be plugged directly into the console.

Loudspeakers: (5x) Genelec 1037C

Five loudspeakers are placed around the hall, resembling a 5.0 surround system as closely as possible. In a true surround system, the center channel (C) is placed directly ahead of the point of capture (0° on-axis). The front left (L) and front right (R) speakers should be placed 22.5° to the left and right of the center speaker, so that there is a 45° arc between the two. Finally, the Left Surround (or Left Side, Ls) and Right Surround (or Right Side, Rs) are symmetrically placed 90-110° to the left and right of the center channel. All speakers should be equidistant from the listener's position.

In this project, the center channel was placed in the direct center of the proscenium – the boundary separating upstage from the downstage apron. The distance between the center channel (C) and the microphone array in the center of the hall measured 37 feet (11.28 meters).

All loudspeakers were carefully positioned to maintain the same distance from the point of capture. At 22.5 degrees, the L and R channels were placed slightly in front of the proscenium, adjacent to the massive columns at the front of the venue. Due to the narrowness of the hall, the Ls and Rs speakers were placed 135° (instead of 110°) in order to maintain congruent distances. Fortunately, this position near the rear columns is also a more probable location for performances featuring musicians behind the audience.

Each speaker was placed on padded chairs in order to minimize mechanical energy transmission from the loudspeaker to the floor. Additional layers of cloth and foam were placed between the chair legs and the floor. This was especially important for the front left and right speakers, as their position overlapped with that of the orchestra pit, which acts as a giant resonating chamber at low frequencies, resulting in an exaggerated bass response. Each speaker was also tilted back and suspended using rope, so that the loudspeaker driver coils were directly on-axis with the microphones in the center of the hall.

Figure 2: Images of Genelec loudspeakers mounted and tilted in padded chairs

Facility: Sursa Performance Hall

This project primarily features Ball State University's Sursa Performance Hall, though the same procedure can be used to record any facility or venue as desired. Sursa Hall has recently celebrated its tenth anniversary; despite this, there is very little documentation of its actual acoustic parameters. Although significant theoretical and mathematical preparation is necessary in acoustic design, there are often too many variables to perfectly predict the final result. After the construction of Sursa Hall was completed, the contractor did not conduct thorough measurements of the actual acoustic characteristics of the hall; after all, it would seem irresponsible to demolish and reconstruct a $20-million dollar facility if the acoustic parameters were not exactly as predicted. The impulse responses recorded over the course of this procedure are readily applicable to statistical analysis of Sursa Hall's sonic properties. Although such dissection exceeds the scope of this project, the impulse responses are nonetheless ready for future analysis in programs such as Rational Acoustic's Smaart v7.0.

Part 3: Software Overview

There is no single infallible technique for recording impulse responses. Many different plugins employ slight variations in sine sweep duration and microphone/loudspeaker configuration. This project will employ the procedures used by three premier convolution reverb plugins: AudioEase's Altiverb, Waves IR1/IR360, and Apple's Logic Pro X's Space Designer utility, which are compatible with mono, stereo, and surround-sound reproduction of a venue. All of these plugins are used by professionals abroad and encourage users to upload and share custom impulse responses.

Altiverb (by Audio Ease)

Altiverb provides instructional resources and materials for recording Altiverb-compatible impulse responses on their website. The steps and procedure described in Part 5 are derived from these resources, available at Audio Ease's website.[15]

Although Altiverb ultimately encodes the impulse responses into a proprietary format usable only to those with the software, it is unique in that this can be done ex post facto. The impulse response measurement procedures for Logic Pro's Space Designer plugin requires that the software be used at the time of recording, as Space Designer simultaneously generates the source signal (sine sweep) and records the impulse response in real-time. Unfortunately, each program prefers sine sweeps of inconsistent duration; it is not possible to record only one sine sweep and import directly into all of the desired formats (Altiverb, Space Designer, Waves, etc.).

Altiverb provides a number of pre-rendered sine sweeps available at the above URL, suitable for a variety of recording formats (44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz) and reverb durations ranging from 3 seconds to 3 minutes. Each of these files begins and ends with a series of noises that identifies the parameters of the sweep, so that Altiverb can process the file accordingly upon import. So long as the recorded impulse responses contain these audio headers and footers, it is possible to record Altiverb-compatible impulse responses using any DAW or recording software. In addition, the impulse responses recorded using the Altiverb test signal can later be processed for further analysis in Smaart 7.0, MaxMSP (utilizing the

[15] Audio Ease, "Making an Altiverb Impulse Response," November 2011, http://www.audioease.com/Pages/Altiverb/sampling.php.

HISSTools_IR_Toolbox and ICST Ambisonics externals), or any other analysis software capable of dual-channel FFT-based audio analysis.

IR-1 and IR-360 (by Waves)

Audio plugins by Waves are some of the finest (and most expensive) in the industry; the brand is internationally renowned by audio professionals. Although there are hundreds of plugins within the Waves library, this project only employs two: IR-1 and IR-360.[16] Both plugins are variations on the same theme of convolution reverb. IR-1 is compatible with mono and stereo formats, while the IR-360 variant is required for high-end surround processing. Like Altiverb, IR-1 and IR-360 can also import impulse responses without needing to operate at the time of recording. However, the sine sweep file provided by Waves is 15 seconds in duration, in contrast to the 30-second sweep utilized by Altiverb. The Waves sweep also lacks the header/footer noises within the .wav file used for identification by the convolution algorithm.

Although the impulse response recording procedures for Waves and Altiverb could be recorded by any DAW or multitrack recording software, the researcher will employ Pro Tools in lieu of Logic due to familiarity.

Space Designer and Impulse Response Utility (Logic Pro X by Apple)

Logic Pro utilizes a different procedure than Altiverb and Waves. The Space Designer reverb plugin includes an Impulse Response utility that acts as a step-by-step template for a variety of microphone/speaker configurations. It is directly compatible with B-Format signals, and includes a specialized procedure for recording a 5.0 surround configuration with both Ambisonic and ORTF techniques. Space Designer is capable of mono-to-stereo conversion, as

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[16] Waves, "IR-1/IR-360 Software Guide," 2013, http://www.waves.com/1lib/pdf/plugins/ir-convolution-reverb.pdf.

well as extrapolation of the B-Format signal to any number of surround formats (4.0, 5.0, 6.0, 7.0, 8.0, etc.).

The simplified procedure described in Part 5 is derived from the Impulse Response Utility instruction manual, available online at Apple's webpage.[17]

Part 4: Equipment Setup

Speaker Playback Configuration

The most basic impulse response measurements are conducted using a single speaker and a minimum of one microphone, usually with an omnidirectional pickup pattern. Although this methodology is able to generate an impulse response that captures the frequency response and reverberation time of the hall, it is only representative of a single *point* source as perceived at only one location. The use of 5 speakers in conjunction with Ambisonic recording technology enables approximation of the venue's acoustic behavior with sounds originating from a variety of locations. By cross-comparing identical test signals projected from identical speakers positioned in a conventional 5.0 surround configuration, it is possible to incorporate parameters of *directionality* and *stereo (spatial) width* to be manipulated in plugin operation. The 5.0 configuration is not applicable only to surround-sound mixing applications; the same setup also includes mono playback (from the center speaker) and stereo (from the front left and right speakers). In post-production, the engineer can specify whether the sound to be reverberated is

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[17] Apple, "Impulse Response Utility – User Manual," 2011, http://help.apple.com/impulseresponseutility/mac/1.0.3/en/impulseresponseutility/usermanual/Impulse%20Response%20Utility%20User%20Manual%20(en).pdf.

narrow and monophonic (such as an instrumental or vocal soloist) or wide and immersive, such as a large ensemble filling the stage or surrounding the audience.

Microphone Placement

The Schoeps ORTF and Soundfield Research MKV Ambisonic microphone are placed on the same microphone stand using a stereo bar. By inverting the Ambisonic microphone, it is possible to position both microphones very closely together while minimizing possible interference due to acute sound shadows caused by XLR cables or stand fixtures. The microphone stand has ample joints that can be manipulated to hoist the array in any direction or angle. The most ideal setup features both microphones arranged in a near-coincident arrangement aligned vertically, so that they are located at the same X and Y coordinates on a three-dimensional plane, with negligible displacement on the Z –axis. This will mitigate noticeable differences in intensity and/or time displacement that might otherwise cause unequal stereophonic bias for one microphone or the other. The boom of the microphone stand can be further manipulated to counteract the effect of the sloped floor.

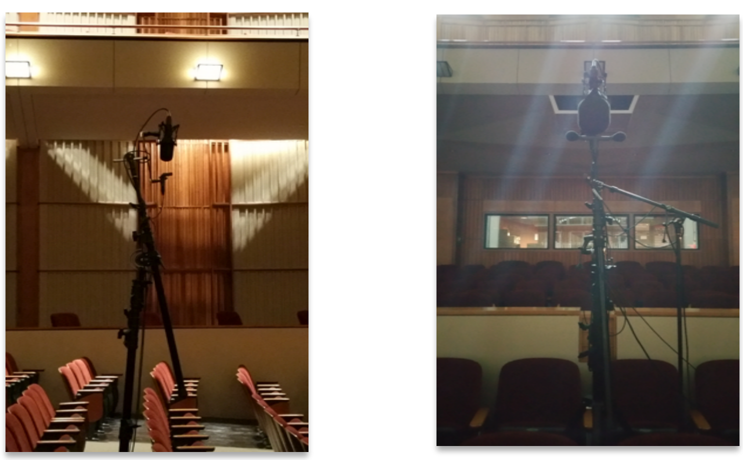

Figure 3: Images of the Soundfield MkV and Schoeps ORTF microphone positioned utilizing the proposed configuration in the center of Sursa Performance Hall.

Recording Equipment Setup

The Schoeps ORTF and SoundField microphones are equipped with proprietary cables. They are not directly compatible with traditional XLR cables, due to the fact that both microphones contain multiple capsules; the Schoeps ORTF is technically two microphones in one fixture, and the SoundField MKV is technically four. Multiple adapters and splitters are required to enable these microphones to cooperate with the traditional XLR inputs on the mixing console and/or audio interface.

The Schoeps 5-Pin cable is split into two XLR inputs, which are connected to the Presonus StudioLive 24.4.2 console. The SoundField MkV microphone cable connects to its corresponding hardware decoder, which converts the 10-pin cable into the B-Format with four outputs (W,X,Y,Z), which are then connected to the Presonus StudioLive 24.4.2 console via four traditional XLR cables.

The Presonus StudioLive 24.4.2 console acts as an audio interface connected to the researcher's MacBook Pro. It allows the researcher to simultaneously send the test signal to the speakers and receive the corresponding response signal from the microphones. The console itself generates a significant amount of ambient noise due to its internal cooling fan. In order to minimize contamination to the impulse response measurements, the recording console and other equipment are positioned within the vestibule to act as an isolated control room. Five channels of audio are sent out from the Presonus StudioLive 24.4.2 console and connected to the CRS Patchbay, which routes the test signal to the 5-speaker array.

Figure 4: The recording equipment is positioned upon a mobile table. The vestibule acts as an isolated control room to isolate the inherent sound noise of the equipment.

Part 5: Impulse Response Recording Procedure

Configure Acoustics in Sursa Hall

Unlike many concert venues, Sursa Performance Hall has variable acoustics: there are seven curtains and ten massive wall panels that can be deployed or stored in order to change the reverberation length and qualities. The impulse response of the hall is directly attributed to the configuration of the curtains and panels at the time of recording. When the absorptive curtains or panels are deployed, they cover reflective surfaces in the hall, which shorts the duration of reverberation. When the curtains and panels are stored, the reflective surfaces beneath are visible, resulting in additional reflections and reverberation. The position of the curtains and panels is controlled by a touch-screen interface, located in a concealed corridor four flights of stairs from the ground floor.

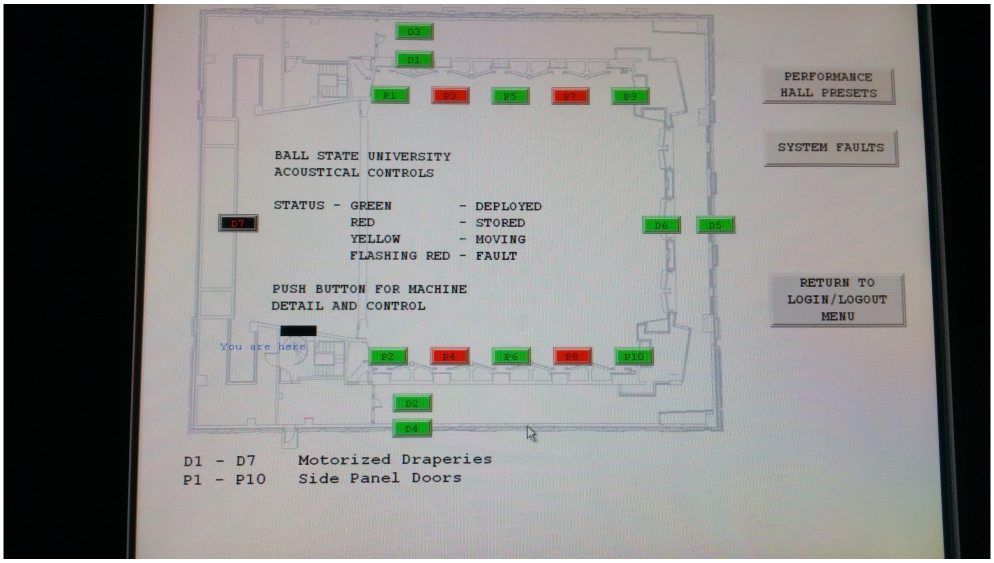

Figure 5: The Touchscreen Control Interface for Sursa Hall's variable acoustic features.

Altiverb and Waves IR-1/IR-360 Procedure (Pro Tools)

- **5.1**: Set curtains/panels as desired.
- **5.2**: Import desired pre-rendered sine sweep test signal from the Altiverb website (30sec sweep for 16sec reverb at 48 kHz or higher).
- **5.3**: Record enable all microphone six channels (two channels from the Schoeps ORTF; four channels from the SoundField MKV in B-Format).
- **5.4:** Play the Altiverb sine sweep (30sec) from only the center channel (mono source; emulating a narrow sound source) and record the playback.
- **5.5**: Play the Altiverb sine sweep (30sec) from only the front left and right speakers (stereo source; emulating a wide ensemble) and record the playback.
- **5.6**: Play the Altiverb sine sweep (30sec) from all five speakers simultaneously (5.0 surround; emulating an immersive ensemble) and record the playback.
- **5.7**: Play the Waves IR-1/360 sine sweep (15sec) from only the center channel (mono) and record the playback.

- **5.8**: Play the Waves IR-1/360 sine sweep (15 sec) from only the front left and right speakers (stereo source; emulating a wide ensemble) and record the playback.
- **5.9**: Play the Waves IR-1 sine sweep (15 sec) from all five speakers simultaneously (5.0 surround; emulating an immersive ensemble) and record the playback.
- **5.10**: Switch to Logic Pro X and perform the Space Designer IR recording procedure while the hall is in the present configuration.

Although the impulse responses recorded using the Altiverb and Waves procedures can be readily imported into Smaart or other programs, the files themselves will not be encoded into their proprietary format until near the end of the project.

Space Designer Recording Procedure (Logic Pro X)

- **5.11**: Configure Impulse Response Utility to record an IR array with a *Stereo* profile and run the program (50 second sweep)
    - 1 Speaker Position (mono, center channel)
    - 2 Mic Positions (Schoeps ORTF, left and right channels)
- **5.12**: Configure Impulse Response Utility to record an IR array with a *True Stereo* profile and run the program (50 second sweep).
    - 2 Speaker Positions (front left and right)
    - 2 Mic Positions (Schoeps ORTF, left and right channels)
- **5.13**: Configure Impulse Response Utility to record an IR array with a *5 Channel B-Format Encoded* profile and run the program (50 second sweep)
    - 5 Speaker Positions (L, C, R, Ls, Rs)
    - 1 Mic Position (Ambisonic: W, X, Y B-Format Signals)

    - Note that Impulse Response Utility discards the Z-signal, which corresponds to the plane parallel to the floor and ceiling.
- **5.14**: Deconvolve and export the Impulse Responses achieved using the above profiles.

Reconfigure Acoustics in Sursa Hall

Once the current configuration of panels and curtains has been satisfactorily captured using all three impulse response measurement procedures, the researcher can prepare for the next batch of measurements. There are thousands of possible acoustic configurations. The hall is "driest" when all curtains and panels are deployed, resulting in maximum absorption and minimal reverb. When all curtains and panels are stored, acoustic reflectivity and reverberation are at a maximum. Once a new configuration has been established, repeat Steps 5.1 – 5.14 to capture sweep responses for the corresponding reverb plugins.

Reposition/Relocate Microphones and Recording Station

The entire procedure thus far has assumed that the microphone stand is positioned at the Front of House console in the center of the hall. If time permits, the microphones should be positioned in other locations, such as on-stage, the front row, the rear of the hall, or even the balcony. The recording console and SoundField decoder are located on a mobile table that can easily be relocated as necessary. Once a new microphone position has been determined, Steps 5.1 – 5.14 should be repeated for every desired acoustic configuration.

Deconvolution and Importing Impulse Responses for Use

After all sweep responses have been recorded and saved as .wav files, they are ready to be imported directly into the corresponding convolution reverb plugins. While the user interface or program dialogue might change depending on the version of the software, in all cases, the reverb plugin itself handles the deconvolution of sweep responses into impulse responses. At this point, the reverb plugin is ready to be used in normal operation. Exact instructions vary on the software and Digital Analog Workstation (DAW) software; software instruction manuals should be consulted for further information.

Part 6: Future Work and Research

Marketing and Dispersal

The final product is a resource *intended for use*. The procedures described above will result in an exhaustive collection of impulse responses that can be used in any software that employs convolution reverb. The plugin allows any musician or audio engineer to synthetically recreate Sursa Hall (and other BSU facilities); if the actual hall is unavailable for a recording session, performers are not denied the advantages of its acoustics properties. The finalized plugins can easily be loaded onto the computers in the MMP studios for student and faculty use.

The methodologies above are modeled after the procedures employed by Altiverb, Space Designer (Logic Pro X), and Waves, all of which are used by professional audio engineers. By adhering to these procedures and encapsulating the impulse responses into a single downloadable plugin, it is possible to submit the Sursa Hall impulse responses to their respective websites for customers to download and use. If intellectual copyright issues permit us to share these plugins, it provides a marketing opportunity for Ball State University and its facilities to audio professionals and amateurs abroad, which is amenable to industry recognition and student recruitment.

Statistical Analysis of Measurements

Utilizing Smaart, FuzzMeasure, or other audio analysis programs, it is possible to perform statistical analyses of the impulse responses. This allows us to measurably quantify the characteristics of our facilities and calculate the differential absorptive/reflective properties of the variable acoustics. After construction was completed in 2004, there was little study and measurement of the hall's parameters. The collection of impulse responses measured by this procedure are ready for import and analysis by future researchers.

- Smaart (by Rational Acoustics)
  - http://www.rationalacoustics.com/store/smaart.html
- FuzzMeasure (by SuperMegaUltraGroovy)
  - http://supermegaultragroovy.com/products/fuzzmeasure/

Uses in MaxMSP Environment

There are a number of external toolkits for the MaxMSP environment that specialize in manipulation of Ambisonic (B-Format) signals and Impulse Responses. A fascinating aspect of SoundField technology is its flexibility after recording has finished; it is possible to "change" the microphone polar pattern, angle, direction, and zoom so long as the neutral W, X, Y, Z channels are preserved. If desired, an engineer can modify the parameters so that the recording sounds like a single cardioid microphone pointed towards the ceiling or the rear of the hall. The SoundField MkV signal can be transformed to simulate a microphone of any type pointing in any direction; it can even any coincident stereo configuration, such as XY or Blumlein. Furthermore, the flexibility of the MaxMSP environment allows for any number of audio channels; it can be used for anything from mono to 10.0 and beyond. The applications are limited only by the creativity of the composer. External toolkits for manipulating impulse responses and ambisonic signals include:

- HISSTools Impulse Response Toolbox for MaxMSP
  - http://eprints.hud.ac.uk/14897/
- Ambisonics Externals for MaxMSP
  - https://www.zhdk.ch/index.php?id=icst_ambisonicsexternals

Conclusion

Much of this project represents the research and development of a comprehensive procedure for recording impulse responses, as well as its implementation utilizing Sursa Performance Hall located at Ball State University. However, the final product also consists of a plethora of audio files to be used in reverb plugins. This paper is accompanied by DVDs containing the raw sweep responses, deconvolved impulse responses, and plugin settings for Altiverb, Waves IR1/360, and Logic Pro X's Space Designer utility. The accompanying materials also include demonstrations of the reverb plugins. If there is difficulty obtaining the data from Ball State University, interested parties may contact the author at nadaywalt@gmail.com.

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